Alvarion Voice Gateways

System Manual

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About This Manual

This manual describes Alvarion's Voice Gateway units and how to install, operate and manage them.

This manual is intended for technicians responsible for installing, setting up and operating the Voice Gateway, and for system administrators responsible for managing the Voice Gateways.

This manual contains the following chapters and appendices:

- **Chapter 1** – **System Description:** Describes the Voice Gateway and its functionality.
- **Chapter 2** – **Installation:** Describes how to install the Voice Gateway and connect it to the SU and to the user's equipment.
- **Chapter 3** – **Using the Web Configuration Server:** Describes how to use the Web Configuration Server for configuring parameters and checking system status.
- **Appendix A – Internal Class 5 Services:** Describes the internal Class-5 services that are supported by the Gateway.
- **Appendix B –** Default Telephony Parameters: Describe the default values for some telephony parameters, including signals/tones parameters, CID parameters and line impedance.
- **Glossary:** Provides definitions of various terms used in the manual.

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Chapter 1 - System Description

In this Chapter:

- Introducing the Voice Gateway, page 2
- Specifications, page 3

1.1 Introducing the Voice Gateway

Alvarion's Voice Gateway enables operators and service providers using Alvarion's Broadband Wireless Access system to provide subscribers with a number of broadband services transparently. The Voice Gateway enables bundling services such as telephony (Voice over IP) and high speed Internet to end-users.

IP-telephony services are supported for standard analog phones or G3 fax machines. The VG-1D1V has a single POTS interface, and the VG-1D2V has two POTS interfaces. The Voice Gateways are available with either H.323 or SIP standard, and support both narrow (compressed) and wideband (uncompressed) speech codecs, silence suppression with comfort noise, line echo cancellation and regional telephone parameters. Class 5 services such call waiting and 3-party conference call are also supported.

Up to 3 telephones can be connected in series to each telephone port. Daisy chaining of Voice Gateways enables the service provider to offer certain end users, for example small offices, additional telephone numbers.

The Voice Gateway also supports Internet access or any other Ethernet based services. The unit can be installed behind a router/NAT due to NAT traversal support allowing signaling as well as voice packets to correctly reach Softswitch or Gatekeeper for bi-directional call initiations. The Gateway can handle up to 16 simultaneous VLANs, enabling the operator to offer different services to different end users behind the unit.

These Gateways incorporate the proprietary DRAP (Dynamic Resources Allocation Protocol) protocol for automatic registration and allocation of resource. DRAP is a protocol based on IP/UDP between the Gateway and a DRAP server (e.g. the BreezeMAX base station). The protocol provides an auto-discovery mechanism for the Gateway, so no specific configuration is required and the Gateway can automatically locate and register with the DRAP server. The protocol uses a few simple messages enabling a Voice Gateway to request resources when calls are made, and the DRAP server to dynamically allocate them.

The Voice Gateways are designed for remote management and supervision using either the built-in internal web server or SNMP.

The Voice Gateways are easily updated and upgraded as they support remote software and configuration file download.

1.2 Specifications

1.2.1 Telephony and Fax Services

Table 1-1: Telephony and Fax Services

Item	Description
VoIP Standard	■ H323 model: H323v2/4 ■ SIP model: SIP (RFC 3261)
Internal Class 5 Services	Call Waiting, 3-party call, call hold and call alteration, differentiated ringing tones (refer to Appendix A for more details)
External Class 5 Services	Activation/deactivation of class 5 services supported by the IP-telephony system
Fax	G3 compliant V.17 14.4 Kbps fax reception and transmission using the T.38 standard (or in-band using G.711 codec)
Calling Number Identification (CNI)	FSK, DTMF
3rd party initiated pause and rerouting	External rerouting of media stream during speech, e.g. for pre-paid calling card and record announcement
DTMF	In-band and out-band using H.245 and H.225
Regional Settings	Telephony signals, tones and cadences (see Default Telephony Parameters, on page 71.

1.2.2 Security

Table 1-2: Security

Item	Description
VLAN	Support IEEE 802.1Q with up to 16 VLAN IDs
Authentication	Per call authentication and registration

1.2.3 Voice Quality

Table 1-3: Voice Quality

Item	Description
Voice Codecs	■ G.711 Ulaw ■ G.711 Alaw ■ G.729ab
Prioritization	■ IEEE 802.1p layer-2 prioritization ■ DiffServ layer-3 prioritization
General	■ Adaptive jitter buffer ■ Echo cancellation ■ Speech sampling rate: 10-60 ms ■ Silence suppression with comfort noise

1.2.4 Configuration and Management

Table 1-4: Configuration and Management

Item	Description
Management Options	■ Internal Web Server ■ SNMP
SNMP Agents	SNMPv1 client MIB II (RFC 1213), Private MIB
Plug & Play Functionality	DHCP, including support messages option 60, 61, 43
Software Upgrade	Using TFTP
Configuration Download	Using TFTP

1.2.5 Bridge Functionality

Table 1-5: Bridge Functionality	
Item	**Description**
Supported Ethernet Devices	Up to 32 MAC addresses
Unknown address Forwarding Policy	Forward Unknown
Bridge Aging Time	180 seconds

1.2.6 Mechanical

Table 1-6: Mechanical Specifications	
Item	**Details**
Dimensions (W x D x H)	17.6 x 11 x 2.8 cm
Weight	1 kg

1.2.7 Electrical

Table 1-7: Electrical Specifications	
Item	**Details**
Power Input	12 VDC from an external power supply, 100-240 VAC, 50-60 Hz, 2A max.
Power Consumption	10.5 W max.

1.2.8 Connectors

Table 1-8: Connectors

Connector	Description	
LAN	Type	10/100Base-TX (RJ-45)
	Cable Length	max 100 m.
PHONE (1 – 2 in VD-1D2V)	Type	RJ-11
	Number of Phones (REN)	Up to 5
	Cable Length	Max. 500 m
WAN	Type	10/100Base-TX (RJ-45)
	Cable Length	max 100 m.
12 VDC	Standard DC power jack to external power supply	

1.2.9 Regulatory Standards Compliance

Table 1-9: Standards Compliance

Type	Standard
EMC	■ Low Voltage Directive (LVD) 73/23/EEC ■ Electromagnetic Compatibility Directive (EMC) 89/336/EEG
Safety	■ IEC 60950 ■ CSA C22.2 No. 950-95/UL 1950 ■ AS/NZS 3260
Emission	■ EN 55022:1998 Class B ■ EN 61000-3-2:1995 ■ Harmonics; EN 61000-3-3:1995 ■ Flicker; FCC part 15 (1998) Class B ■ AS/NZS 3548 (1995)
Immunity	EN 55024:1998

1.2.10 Environmental

Table 1-10: Environmental Specifications

Item	Details
Operating temperature	0 °C to 50 °C
Operating humidity	10%-95% RH non condensing

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Chapter 2 - Installation

In this Chapter:

- Installation Requirements, page 10
- Front and Rear Panel Components, page 11
- Installation, page 13

2.1 Installation Requirements

2.1.1 Packing List

- Voice Gateway with one (VG-1D1V) or two (VG-1D2V) Phone Ports
- Power supply with a DC connecting cable
- Mains power cable

2.1.2 Additional Installation Requirements

- A straight Ethernet cable for connecting the WAN port to the SU-IDU
- An Ethernet cable for connecting to the user's data equipment (straight for connecting to a PC, crossed for connecting to a hub/switch)
- Standard phone cable(s) with RJ-11 connectors.
- Mains plug adapter (if the power plug on the supplied mains power cable does not fit local power outlets).
- Portable PC with an Ethernet card and an Ethernet cable for configuring the Voice Gateway parameters using a web browser.

2.2 Front and Rear Panel Components

2.2.1 Connectors

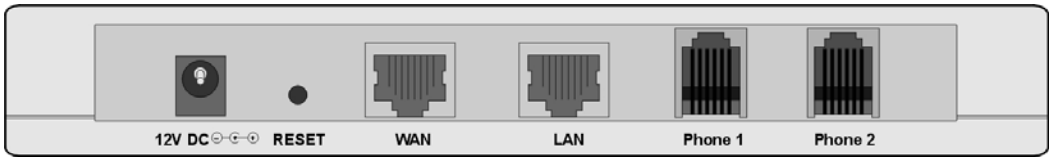

Figure 2-1: Voice Gateway VG-1D2V Back Panel

NOTE

The VG-1D1V has a single Phone connector.

Table 2-1: Voice Gateway Connectors

Name	Connector	Functionality
Phone 1	RJ-11	Connections to the user's telephones
Phone 2 (VG-1D2V only)	RJ-11	Connections to the user's telephones
LAN	10/100Base-T (RJ-45)	Connection to the user's data equipment
WAN	10/100Base-T (RJ-45)	Connection to the SU-IDU
12 VDC	DC power jack	Connection to power supply

2.2.2 Reset to Factory Default Configuration

Press down the RESET button on the back of the unit for at least 5 seconds to reset all configurable parameters back to their original default values. After releasing the RESET button, the PWR, WAN and LAN LEDs bilnk twice, indicating proper operation. The affect on the selected IP parameters acquisition method depends on the time the RESET button is held in the pressed position:

- If the RESET button is pressed down for 5 to 10 seconds: The unit will use DHCP to get the IP parameters.
- If the RESET button is pressed down for more than 10 seconds: The unit will use the static (manually defined) IP parameters.

For more details on configuration of DHCP and static IP parameters, refer to WAN Configuration Page on page 23.

2.2.3 LEDs

Figure 2-2: VG-1D2V Front Panel

NOTE

The VG-1D1V has a single Phone LED.

Table 2-2: Voice Gateway LEDs

Name	Symbol	Description	Functionality
Phone 1		Phone service indication	■ Off –Phone line does not get IP telephony services ■ On – Phone line is connected to the IP-telephony system
Phone 2 (VG-1D2V only)		Phone service indication	■ Off –Phone line does not get IP telephony services ■ On – Phone line is connected to the IP-telephony system
LAN		LAN port status indication	■ Off – Ethernet Link not detected ■ On – Ethernet link connected, no activity ■ Blinking – Ethernet link activity
WAN		WAN port status indication	■ Off – Ethernet link not detected ■ On – Ethernet link connected, no activity ■ Blinking – Ethernet link activity
POWER	PWR	Power Indication	■ Off – unit is not powered or power failed ■ Green – power OK

2.3 Installation

The unit can be placed on a desktop or a shelf. The location should be selected taking into account the necessary connections to mains power, SU-IDU and user's data/telephony equipment.

To install the Voice Gateway:

1. Use a straight Ethernet cable to connect the WAN port on the rear panel of the unit to the Ethernet port of the SU-IDU. The length of the cable, together with the length of the cable connecting the SU-IDU to the SU-ODU, should not exceed 100m.
2. Connect the DC power cable of power supply to the 12 VDC jack on the rear panel of the unit.
3. Connect the mains power cable to the power supply. Connect the other end of the mains power cable to the AC mains.

NOTE

The color codes of the power cable are as follows:

Brown	Phase	~
Blue	Neutral	0
Yellow/Green	Ground	⏚

4. After power up, all fron panel LEDs bilnk once, and then the PWR, WAN and LAN LEDs bilnk twice, indicating that the unit operates properly. Then the PWR LED is lit. Other LEDs may also be lit, according to the status of the WAN, LAN and Phone ports, as described in section 2.2.3.
5. Connect a PC to the Ethernet port using a straight Ethernet cable, and use a web browser to configure the parameters of the Voice Gateway. See Chapter 3 - Using the Web Configuration Server for details.

NOTE

To enable remote management of the Voice Gateway, the Management & voice VLAN ID must be configured.

6. Connect the 10/100 Base-T Ethernet connector to the data equipment. The length of the Ethernet cable should not exceed 100m. Use a straight cable for connecting to a PC, or a crossed cable for connecting to a hub/switch)
7. Use standard telephone cord(s) with RJ-11 termination to connect the telephony equipment to the unit.
8. If needed, configure the basic parameters of the SU and allign its antenna for optimal performance. For details refer to the applicable System Manual.

9 Verify that the SU is connected to the base station.

10 To verify data connectivity, from the end-user's PC or from a portable PC connected to the unit, try to connect to the Internet or to ping another unit in the network.

11 Verify proper operation using the LED indicators (see Table 2-2).

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Chapter 3 - Using the Web Configuration Server

In this Chapter

- Introduction to the Web Configuration Server, page 16
- Accessing the Web Configuration Server, page 17
- Using the Web Configuration Server, page 18
- Home Menu - Product info Page, page19
- WAN Menu, page 21
- VLAN Tagging Menu, page 26
- Telephone Menu – SIP Model, page 34
- BW Reservation – DRAP Configuration Page, page 47
- System Menu, 51
- Upgrade Page, page 55
- Restart Page, page 57

3.1 Introduction to the Web Configuration Server

The Voice Gateway can be configured using the following methods:

- The Web Configuration Server
- An .ini-file loaded into the unit from a TFTP-server

This document describes the configuration using the Web Configuration Server.

3.2 Accessing the Web Configuration Server

Follow the steps below to access the Web Configuration Server:

1 Connect the unit to the AC mains and to the SU-IDU.

2 If a DHCP server is being used, the unit may request an IP address during the power up (depending on the .ini file in the unit).

3 If fixed IP address should be used, proceed as follows:

- ◇ Press the RESET button on the back of the unit
- ◇ Keep the RESET button pressed for more than 10 seconds. Make sure that the unit reboots properly (all LEDs on the front panel will blink once, and then the PWR, WAN and LAN LEDS blink twice)
- ◇ Release the RESET button. The PWR, WAN and LAN LEDs should blink twice, indicating proper operation of the unit.

After this sequence the Voice Gateway will be at "factory default" status, with IP address 192.168.254.254 and subnet mask 255.255.255.0.

4 Open a web browser (Internet Explorer 5.5 or higher).

NOTE

Be sure to disable caching of web pages.

5 Enter the IP address of the unit in the address field.

6 If the Web Configuration Server is password protected, you will be prompted to enter your password in order to login to the system. The default password is **installer** .

7 The Web Configuration Server main view appears on the screen.

3.3 Using the Web Configuration Server

The Web Configuration Server view consists of a number of menu links (to the left). Clicking on each of them will display the configuration/status page for the selected menu item, with the applicable content (configurable parameters/options or status information) in the main area. Several pages include a page selection bar at the top of the page, enabling selection between several pages related to the same menu item.

Figure 3-1: Web Configuration Page

CAUTION

Many pages include a "Save Settings" button. Click on the Save Settings button before selecting another page/menu item, or before quitting the application. The Save Settings functionality in many cases is per page – if you leave the page without clicking the Save Settings button, all the changes in the page will be lost.

Changes to most of the settings are applied only after restarting the unit (refer to section 3.11).

CAUTION

There is no control that the entered values are valid or have the correct format or range. If invalid values are entered, access to the unit may be lost and in that case a factory default procedure must be performed. Refer to section 2.2.2 for information about how to reset the Voice Gateway to factory default parameters.

3.4 Home Menu - Product info Page

The Product Info page provides general information on the Voice Gateway.

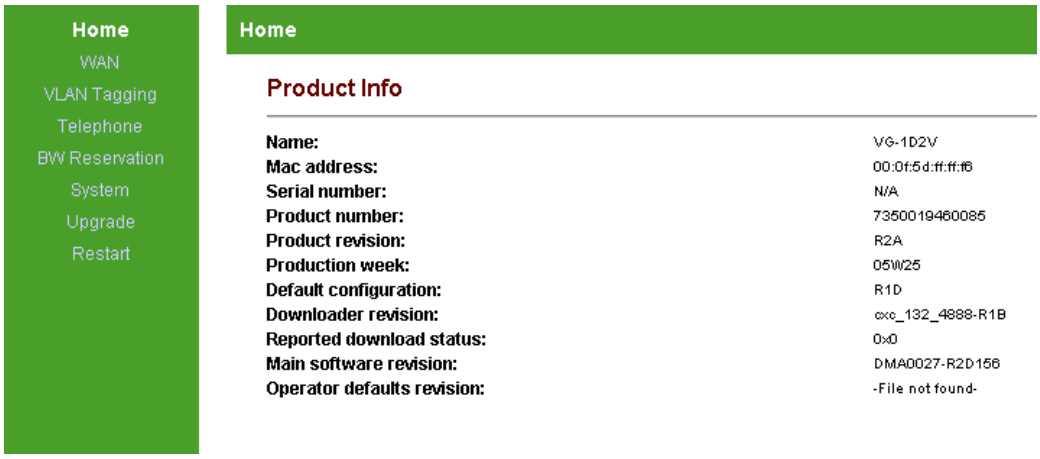

Figure 3-2: Product Info Page

The Product Info page includes the following components:

Table 3-1: Product Info Page Parameters

Parameter	Description
Name	The unit's model
Mac address	The MAC address of the unit
Serial Number	The serial number of the unit
Product number	Not Used
Product revision	The hardware revision
Production week	Production date in the format <yy>w<ww>. <yy> is the year (two last digits) and ww is the week (two digits).
Default configuration	The unit's configuration
Downloader revision	The revision of the SW download SW module.
Reported download status	The status of the SW download operation. For more details refer to Downloader Result Codes (hexadecimal) on page 55.
Main software revision	The unit's main SW version
Operator defaults revision	The custom .ini file (if exists)

In any case of contact with Alvarion Customer Service, include the Default configuration, Downloader revision, Main software revision and Operator defaults revision (.ini file) if exists.

3.5 WAN Menu

The WAN menu page includes settings related to the operation and functionality on the WAN (network) side of the unit.

NOTE

Be careful when setting these parameters to avoid conflicts in the network.

The WAN page selection bar includes the following options:

- WAN Status (WAN Status Page)
- WAN Configuration (WAN Configuration Page)

3.5.1 WAN Status Page

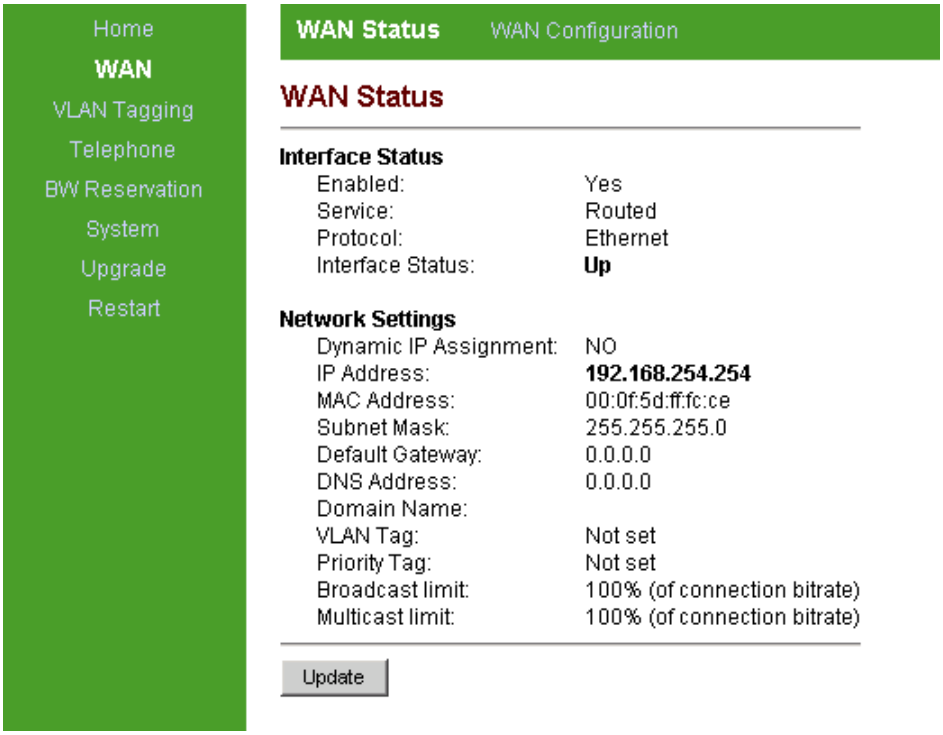

Figure 3-3: WAN Status Page

The WAN Status page includes the following components:

Table 3-2: WAN Status Page Parameters

Parameter	Description
Interface Status	
Enabled	The administrative status of the WAN port: Yes or No. In current version the administrative status cannot be disabled.
Service	The configured operation mode. In current version it is always Bridge.
Protocol	The protocol used for data transmission: Ethernet or PPPoE
Interface Status	The operational status of the WAN port: Up or Down.
Network Settings	
Dynamic IP Assignment	The method of configuring IP Address, Subnet Mask, Default Gateway and DNS Address, as defined in the WAN Configuration page: ■ Yes (via DHCP): the parameters are obtained from a DHCP server. ■ No: the parameters are configured manually
IP Address	The IP address of the unit
MAC Address	The MAC address of the unit
Subnet Mask	The IP Subnet Mask
Default Gateway	The Default Gateway address
DNS Address	IP DNS Server address
Domain Name	The Domain Name as defined in the WAN Configuration page
VLAN Tag	The VLAN ID tag defined for management traffic
Priority Tag	The Priority tag defined for management traffic
Broadcast Limit	The upper limit on the bit rate of broadcast packets that will be forwarded as a percentage of the incoming traffic bit rate, as defined in the WAN Configuration page.
Multicast Limit	The upper limit on the bit rate of broadcast packets that will be forwarded as a percentage of the incoming traffic bit rate, as defined in the WAN Configuration page.

Click on the **Update** button to refresh the display.

3.5.2 WAN Configuration Page

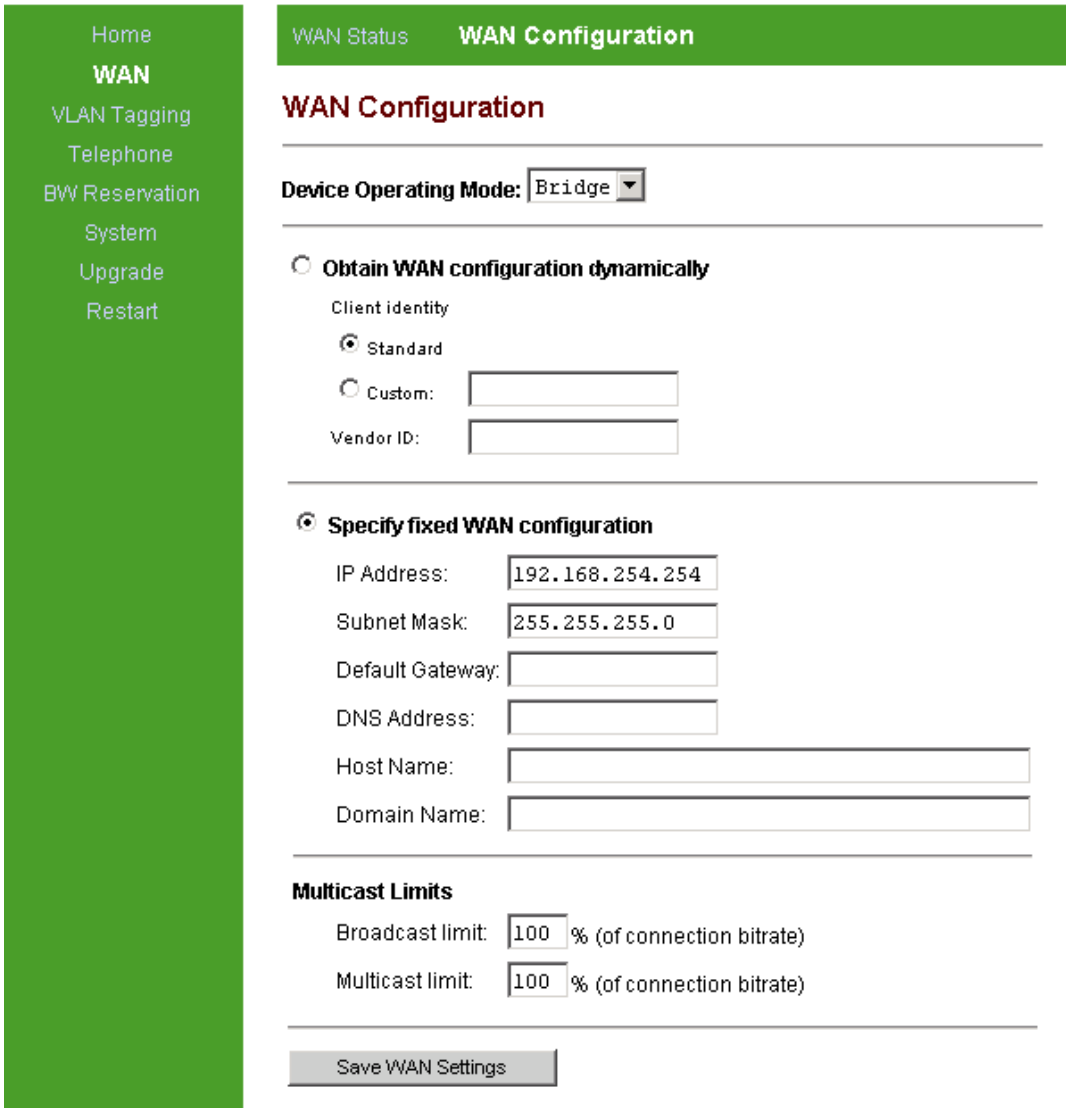

Figure 3-4: WAN Configuration Page

The WAN Configuration page includes the following components:

Table 3-3: WAN Configuration Page Parameters

Parameter	Description
Device Operating Mode	The operating mode of the unit. In current version the operation mode is always Bridge.
Obtain WAN configuration dynamically	Select this option to obtain IP parameters from a DHCP server. The default is that the option is selected.

Table 3-3: WAN Configuration Page Parameters

Parameter	Description
Client identity	Applicable only if the "Obtain WAN configuration dynamically" option is selected. The method used for identifying the client (Option 61). The options are: ■ **Standard**: The unit's MAC address ■ **Custom**: An identification string of up to 25 characters. The default is null (an empty string)
Vendor ID	Applicable only if the "Obtain WAN configuration dynamically" option is selected. The Vendor ID (Option 60). A string of up to 25 characters. The default used by the unit is VoIP (not displayed).
Specify fixed WAN configuration	Select this option to configure the IP parameters manually. The default is that the option is not selected (the Obtain WAN configuration dynamically option is selected).
IP Address	Applicable only if the "Specify fixed WAN configuration" option is selected. The IP address of the unit. The default is 192.168.254.254
Subnet Mask	Applicable only if the "Specify fixed WAN configuration" option is selected. The IP Subnet Mask. The default is 255.255.255.0
Default Gateway	Applicable only if the "Specify fixed WAN configuration" option is selected. The Default Gateway address. The default is none (empty)
DNS Address	Applicable only if the "Specify fixed WAN configuration" option is selected. IP DNS Server address. The default is none (empty)
Host Name	The Host name for clients. A string of up to 25 characters. The default is null (an empty string).
Domain Name	The Domain Name for client resolution. A string of up to 25 characters. The default is null (an empty string).
Broadcast Limit	The upper limit on the bit rate of broadcast packets that will be forwarded as a percentage of the incoming traffic bit rate. The default is 100%.

Table 3-3: WAN Configuration Page Parameters	
Parameter	**Description**
Multicast Limit	The upper limit on the bit rate of broadcast packets that will be forwarded as a percentage of the incoming traffic bit rate. The default is 100%.

Click on the **Save WAN Settings** button before leaving the page to save the new settings. The new settings will be applied after restarting the unit.

3.6 VLAN Tagging Menu

The VLAN Tagging page selection bar includes the following options:

- VLAN Tagging (VLAN Tagging Page)
- VoIP VLAN Configuration (VoIP VLAN Configuration Page)

3.6.1 VLAN Tagging Page

The Voice Gateway supports 802.1Q VLAN standard, allowing IEEE 802 Local Area Networks (LANs) of all types to be connected together with Media Access Control (MAC) Bridges, as specified in ISO/IEC 15802-3. This standard defines the operation of Virtual LAN (VLAN) Bridges that permit the definition, operation and administration of Virtual LAN topologies within a bridged LAN infrastructure.

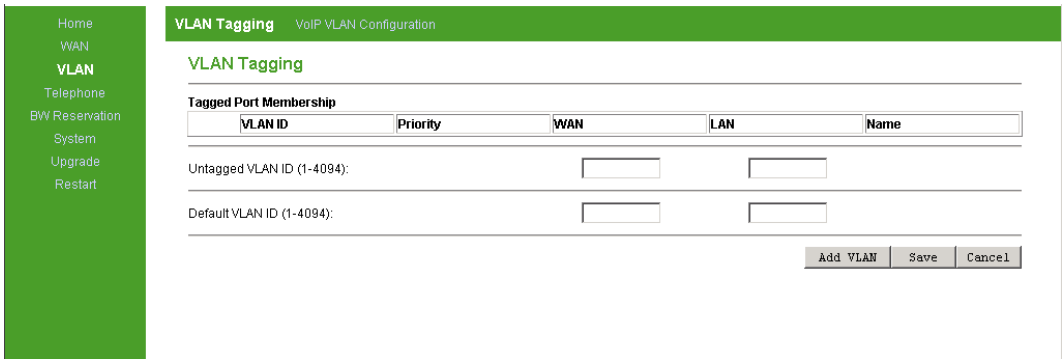

Figure 3-5: VLAN Tagging Page

The VLAN page enables defining up to 16 VLANs, and it includes the following components:

Table 3-4: VLAN Page Parameters

Parameter	Description
Tagged Port Membership	A table displaying the defined VLANs. For details on modifying the table refer to section 3.6.2 below.
Untagged VLAN ID	The VLAN ID that is defined for untagged data on the WAN port (text box on the left side) and the LAN port (text box on the right side). This is the VLAN ID that will be used internally for incoming data. The range for both parameters is from 1 to 4094.
Default VLAN ID	The text box on the left side is for the WAN port. This is the VLAN defined for management frames (SNMP, HTTP, TFTP) arriving on the WAN port. The text box on the right side is for the LAN port. Typically this field should remain empty. It is used to specify the VLAN for non-VoIP traffic (RTP, Call Signaling) arriving on the LAN side when Voice Gateways are daisy-chained. The range for both parameters is from 1 to 4094.

3.6.2 Adding and Deleting VLANs

To add a VLAN:

1 Click on the Add VLAN button. The VLAN Editor (Add) is displayed:

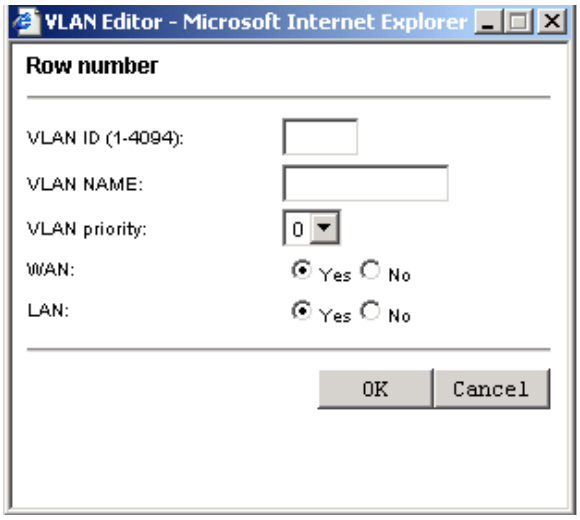

Figure 3-6: VLAN Editor (Add VLAN)

2 Enter the VLAN ID (1 to 4094), VLAN NAME (A descriptive string of printable characters. Do not use special characters such as space or comma), and the VLAN priority tag (0 to 7).

3 If applicable packets need to be tagged on the WAN/LAN port, check the relevant Yes option. Otherwise check the No option. Note that only one VLAN can be untagged on each port (or on both).

4 Click OK. The newly added entry will be added to the Tagged Port Membership table.

To delete a VLAN from the Tagged Port Membership table:

1 Click on the row ID number of the entry you wish to remove. The VLAN Editor (Delete) is displayed:

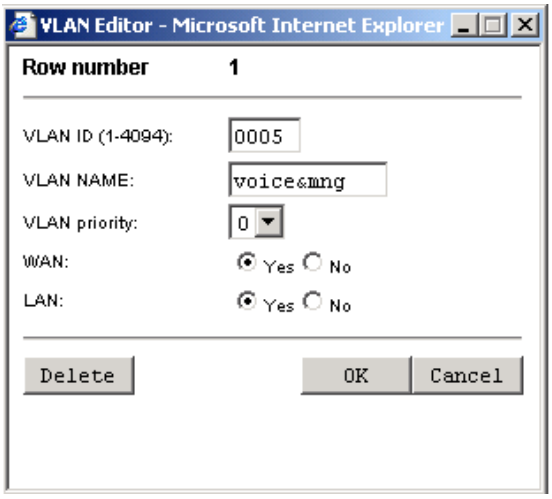

Figure 3-7: VLAN Editor (Delete VLAN)

2 Click on the Delete button. The entry will be removed from the Tagged Port Membership table.

3.6.3 VoIP VLAN Configuration Page

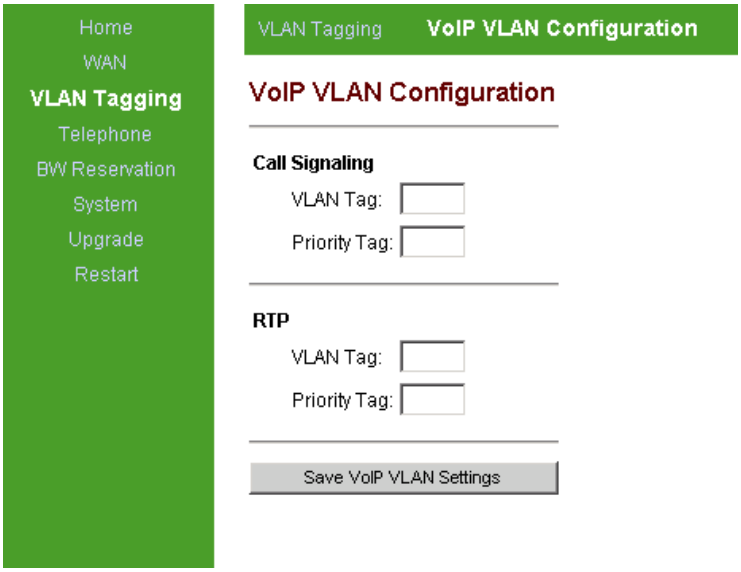

Figure 3-8: VoIP VLAN Configuration Page

The VoIP VLAN configuration page enables defining the following parameters:

Table 3-5: VoIp VLAN Configuration Page Parameters

Parameter	Description
Call Signaling	
VLAN Tag	The VLAN ID tag for VoIP call signaling packets
Priority Tag	The Priority tag for VoIP call signaling packets
RTP	
VLAN Tag	The VLAN ID tag for RTP and RTCP packets
Priority Tag	The Priority tag for RTP and RTCP packets

Typically, the same VLAN is used for management, call signaling and RTP. In this case, the same VLAN and Priority Tags should be configured for management (Default VLAN on WAN port in the VLAN Tagging page), Call Signaling and RTP. However, the Voice Gateway supports separation of VLANs and allows defining 3 different VLANs for management, call signaling and RTP traffic.

3.6.4 VLAN Configuration Example 1

This example describes how to define the following configuration:

- VLAN ID 100, VLAN Priority 7 for Voice (call signaling, RTP and RTCP) and Management packets on the WAN port.
- No VLAN for data on the LAN and WAN ports.

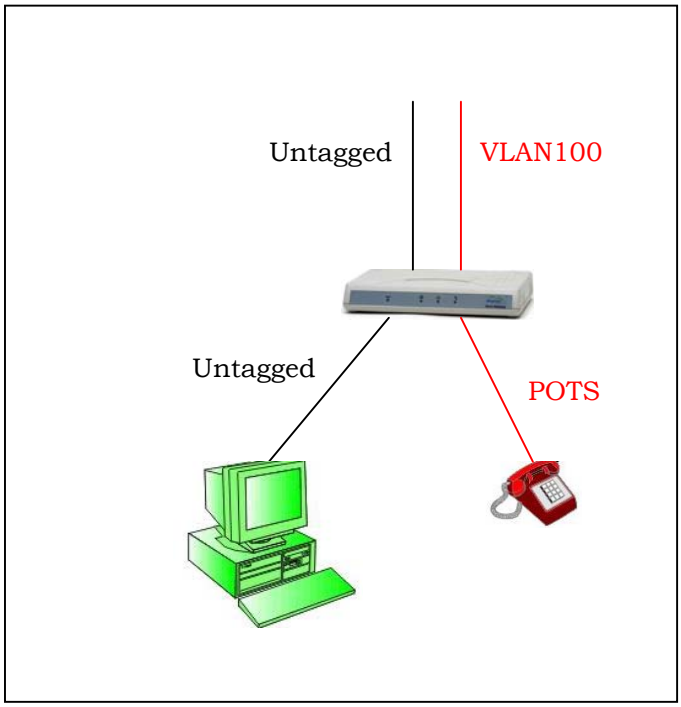

Figure 3-9: VLAN Configuration Example 1

1. In the VLAN page, click Add VLAN to open the VLAN Editor.
2. In the VLAN Editor, enter the follwing for Voice and Management VLAN:
   - VLAN ID: 100
   - VLAN NAME: Voice&Mng
   - VLAN Priority: 7
   - WAN: Yes
   - LAN: No
3. Click OK to add the VLAN to the Tagged Port Membership table.

4 Enter the VLAN ID for Voice and Management (100) in the field Default VLAN ID on WAN port, and click Save.

5 In the Page Selection bar, click on VoIP VLAN Configuration to open the VoIP VLAN Configuration page. Enter 100 in the VLAN Tag fields for both Call Signaling and RTP. Enter 7 in the Priority Tag field for both Call Signaling and RTP. Click Save VoIP VLAN Settings. Go back to the VLAN Tagging page.

6 In the VLAN page, click Add VLAN to open the VLAN Editor.

7 In the VLAN Editor, enter the follwing for untagged data:

- VLAN ID: 200 (an arbitrary selection-a VLAN ID is required for defining the untagged data. This VLAN tag is only used internally in the unit)
- VLAN NAME: Untagged
- VLAN Priority: 0
- WAN: No
- LAN: No

8 Click OK to add the VLAN to the Tagged Port Membership table.

9 Enter the VLAN ID for untagged data (200) in the fields Untagged VLAN ID on LAN port and Untagged VLAN ID on WAN port, and click Save.

10 Restart the unit to apply the changes.

3.6.5 VLAN Configuration Example 2

This example describes how to define the following configuration:

- VLAN ID 100, VLAN Priority 7 for Voice (call signaling, RTP and RTCP) and Management packets on the WAN port.
- Two types of data traffic:
  - Data with VLAN ID 200, Priority 3 on the LAN and WAN ports
  - Untagged (no VLAN) data on the LAN and WAN ports.

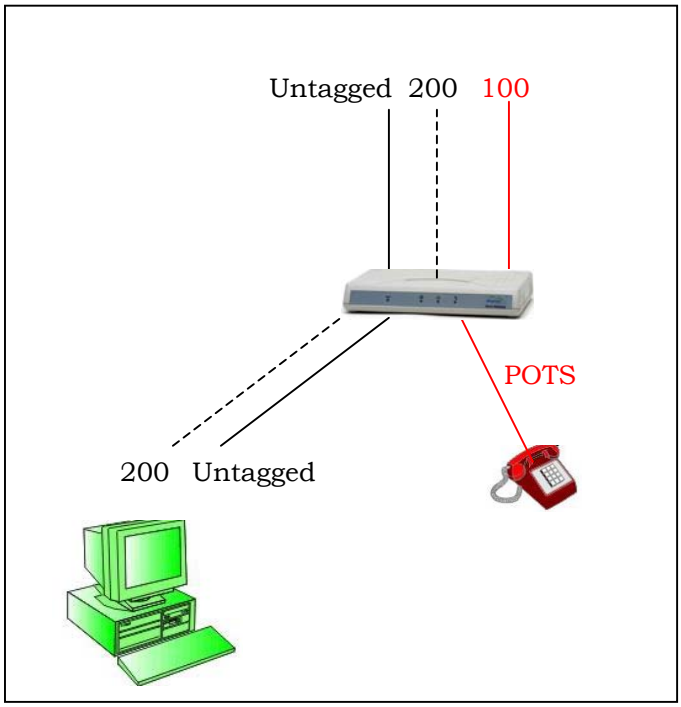

Figure 3-10: VLAN Configuration Example 2

1. In the VLAN page, click Add VLAN to open the VLAN Editor.
2. In the VLAN Editor, enter the follwing for Voice and Management VLAN:
   - VLAN ID: 100
   - VLAN NAME: Voice&Mng
   - VLAN Priority: 7
   - WAN: Yes

- LAN: No

3. Click OK to add the VLAN to the Tagged Port Membership table.
4. Enter the VLAN ID for Voice and Management (100) in the field Default VLAN ID on WAN port, and click Save.
5. In the Page Selection bar, click on VoIP VLAN Configuration to open the VoIP VLAN Configuration page. Enter 100 in the VLAN Tag fields for both Call Signaling and RTP. Enter 7 in the Priority Tag field for both Call Signaling and RTP. Click Save VoIP VLAN Settings. Go back to the VLAN Tagging page.
6. In the VLAN page, click Add VLAN to open the VLAN Editor.
7. In the VLAN Editor, enter the follwing for Data with VLAN 200:

   - VLAN ID: 200
   - VLAN NAME: DATA_200
   - VLAN Priority: 3
   - WAN: Yes
   - LAN: Yes

8. Click OK to add the VLAN to the Tagged Port Membership table.
9. In the VLAN Editor, enter the follwing for untagged data:

   - VLAN ID: 1000 (an arbitrary selection-a VLAN ID is required for defining the untagged data. This VLAN tag is only used internally in the unit)
   - VLAN NAME: Untagged
   - VLAN Priority: 0
   - WAN: No
   - LAN: No

10. Click OK to add the VLAN to the Tagged Port Membership table.
11. Enter the VLAN ID for untagged data (200) in the fields Untagged VLAN ID on LAN port and Untagged VLAN ID on WAN port, and click Save.
12. Restart the unit to apply the changes.

3.7 Telephone Menu – SIP Model

The Telephone (SIP Model) page selection bar includes the following options:

- SIP (SIP Configuration Page)
- SIP Extensions (SIP Extensions Page)
- NAT (NAT Traversal Configuration Page)
- ToS/DiffServ (ToS/DiffServ Page)

3.7.1 SIP Configuration Page

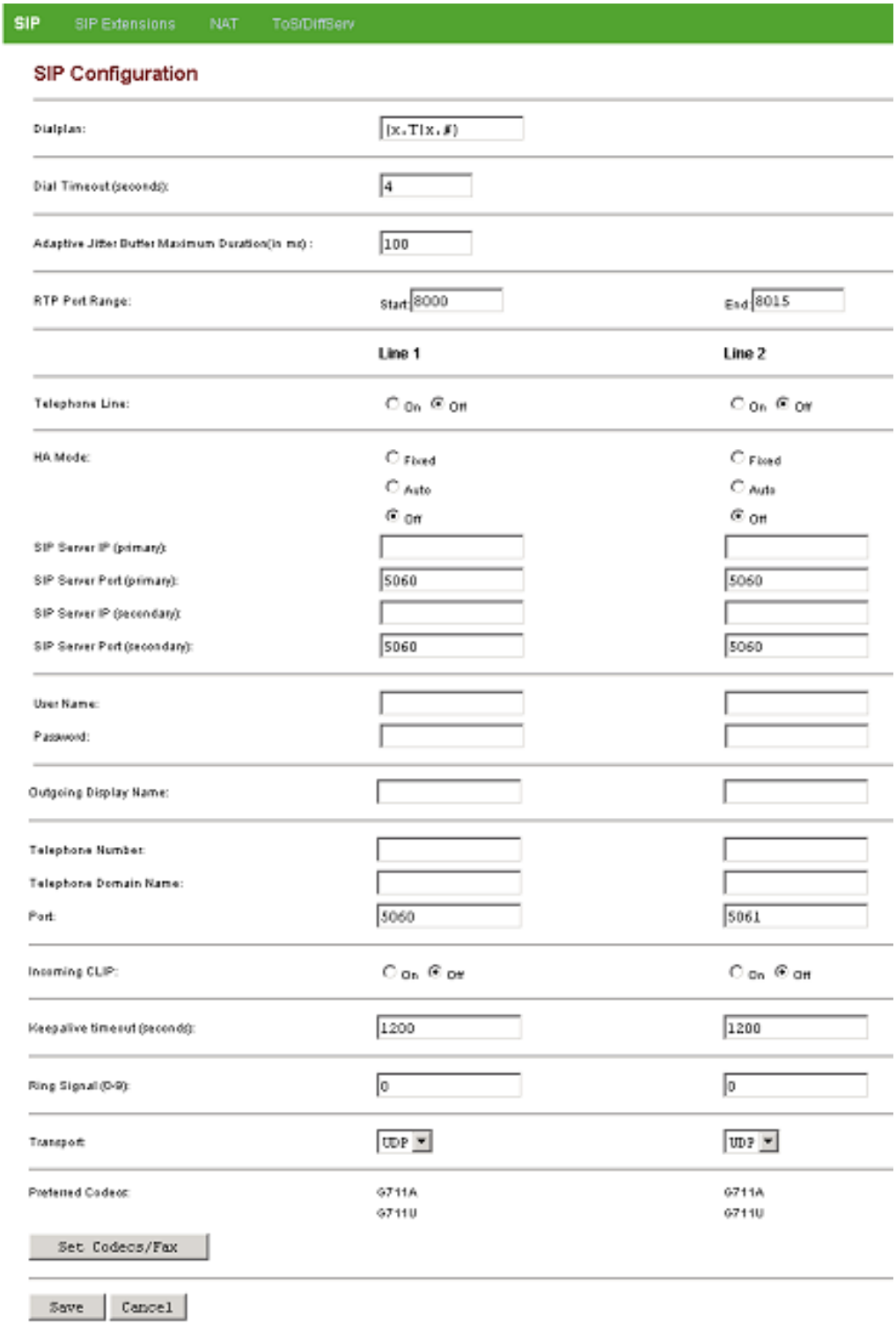

Figure 3-11: SIP Configuration Page (VG-1D2V)

The SIP Configuration page includes the following components:

Table 3-6: SIP Configuration Page Parameters

Parameter	Description
Dialplan	The Dialplan parameter defines how the Voice Gateway decides that a complete number has been dialed. See more details in section 3.7.1.2 on page 41. The default value is x.T|x.#, which means that each of the following schemes can be used: ■ x.T: Dial timeout. Any number of digits may be dialed. Following T seconds in which no new digit is dialed, a decision is reached that dialing was completed and the unit will send the dialing sequence received up to this time as a complete telephone number. This is necessary since the whole telephone number is sent at once and not digit by digit. ■ x.#: Any number of digits may be dialed. A decision that dialing was completed will be reached once # is pressed. The combination of both schemes means that dialing is completed either after a timeout of T seconds or after pressing #.
Dial timeout	The timeout in seconds for the dial timeout dialplan. The number of seconds that the unit waits before it sends a complete telephone number. This is necessary since the whole telephone number is sent at once and not digit by digit. The range is 1 to 60 seconds Default value is 4 seconds.
Adaptive Jitter Buffer Maximum Duration	The Voice Gateway uses a Jitter Buffer to eliminate jitter effects. The size of the buffer changes dynamically to reflect actual jitter conditions. The Adaptive Jitter Buffer Maximum Duration defines the maximum size that is available for the jitter buffer (the higher the size, the higher the potential delay). The range is 100 to 300 milliseconds. The default duration is 100 milliseconds.
RTP Port Range	The start and end UDP port-range for RTP protocol. Recommended values for Start and End ports are in the range 1030-65535. The default Start port is 8000. The default End port is 8015.
Telephone line	Switch the telephone line On or Off. The default is Off.

Table 3-6: SIP Configuration Page Parameters

Parameter	Description
HA mode	The High Availability mode defines the support of a secondary Gate keeper/SIP Server for high system availability, redundancy, and scalability. When a secondary server is available, the unit will try to register to the secondary server after 10 failed attempts to register to the primary server. The available options are: ■ Fixed: The secondary SIP Server IP address is defined manually by the SIP Server IP (secondary) parameter. ■ Auto: The secondary SIP Server/Proxy IP address is supplied by the primary SIP Server. ■ Off: Secondary Gate keeper/SIP Server is not supported. The default is Off.
SIP Server IP (primary)	The IP address for the primary SIP server/proxy who is responsible for managing the Voice Gateway in the specific network. If HA-mode is set to Auto, the primary SIP server/proxy provides to the Voice Gateway during registration an IP-address for the secondary system.
SIP Server Port (primary)	The port used for the primary system. The recommended values are in the range 1030-65536. The default is 5060.
SIP Server IP (secondary)	The IP address of the secondary SIP server/proxy.
SIP Server Port (secondary)	The port used for the secondary system. The recommended values are in the range 1030-65536. The default is 5060.
User Name	The SIP user Name. Format (name or number) depends on the SIP server. A string of up to 25 characters.
Password	The SIP user Password. Format (name or number) depends on the SIP server. A string of up to 25 characters.
Outgoing Display name	The name to be displayed on the caller ID display of a receiving party (if supported by the network). Up to 25 characters.

Table 3-6: SIP Configuration Page Parameters

Parameter	Description
Telephone number	The telephone number of the specific telephone line to be used when registering the unit at the Gate keeper/SIP Server. The telephone number is limited to 25 characters. It may also be an e-mail address (limited to 25 characters before the @ sign). The Telephone number must be set to a unique value for each telephone line in the network in order for the system to accept it.
Telephone domain name	The domain-name. The Telephone domain name is limited to 25 characters, i.e. 25 characters after the @-sign. If not specified by the user, the same information as defined in the SIP Server IP field will be used.
Port	The number of the outgoing signaling port on the telephone line. Line1 and Line2 cannot have the same port number. The range is from 1030 to 65535. The default is 5060 for Line 1 and 5061 for Line 2.
Incoming CLIP	The Calling Line Identity Presentation (Caller ID) option for the telephone line. If On is selected, the Caller ID information of a calling party in incoming calls will be displayed on a caller ID display attached to the telephone line. Caller ID method should be configured by loading an appropriate .ini file (refer to the Upgrade section for information). The default is Off.
Keepalive timeout (seconds)	The interval suggested by the Voice Gateway for sending the keep alive messages to the network. If Keep-alive timeout is sent from the network, it will override the setting in the Voice Gateway. If the unit does not succeed to renew registration after half of the timeout (600 seconds with the default timeout of 1200 seconds), it will continue the registration attempts, but the Phone ports will be disabled until successful registration: no dial tone, Phone LEDs are off. The range is from 10 to 65535 seconds. The default is 1200 seconds.

Table 3-6: SIP Configuration Page Parameters

Parameter	Description
Ring signal [0 – 9]	The Ring signal parameter provides a selection of 10 different ring patterns (0-9) that the unit can use. The default is 0.
Transport	Configure whether signaling shall use UDP or TCP. The default is UDP.
Preferred codecs	Displays the currently supported codecs, according to the defined priorities. Click the Set Codecs/Fax button to change codecs settings/priorities. NOTE: Click Save before clicking the Set Codecs/Fax button. Otherwise, all configuration changes in the Telephone page will be lost.

Click on the **Save** button before leaving the page to save the new settings. The new settings will be applied after restarting the unit.

Click the **Set Codecs/Fax** button to change change codecs settings/priorities as described below.

3.7.1.1 Codecs and Fax Configuration

After clicking the Set Codecs/Fax button, the Codecs and Fax Configuration page is displayed.

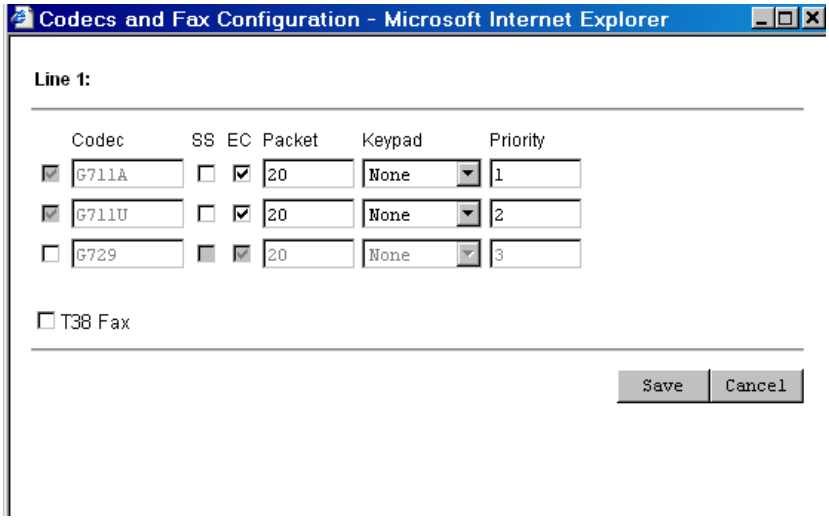

Figure 3-12: Codecs and Fax Configuration Window – VG-1D1V

The following settings are available for each line:

Table 3-7Codecs and Fax Configuration Page Parameters

Parameter	Description
Codec	The Codec check boxes identify which codecs are used. G.711A and G711U are mandatory and cannot be deselected. G.729 is optional. The default is G729 codec selected (checked). NOTE: G 729 with Annex A is implemented in the Voice Gateway. It enables communication with devices using either G729 with Annex A or G729 with Annex A and Annex B. It is not possible to communicate with devices using G729 with Annex B only. For each Codec in use, the following can be configured:
SS	The SS (Silent Suppression) option for outgoing calls. When the SS option is enabled, silence intervals are identified and only relevant information is transmitted, using less bandwidth than during voice activity intervals. This allows for a better overall utilization of the available bandwidth. It is possible to enable Silent Suppression with G729 codec. Silent Suppression is not applicable when using the G711 codecs. The default (G729) is SS disabled.
EC	The EC (Echo Cancellation) option, defines whether to activate the echo cancellation mechanism for improved voice quality. EC is not used during Fax (T.38) transmissions. The default is enabled.
Packet	The packet size in milliseconds. The range is from 10 to 150 milliseconds. The default is 30 ms for G729 and 20 ms for G711A and G711U.
Keypad	The "Keypad" field indicated which transmission method to be used for user input DTMF signaling (i.e. phone banking). "None" means in-band, which should be used with G.711 only. The options are None, RFC2833 and SIP INFO. RFC2833 and SIP INFO should be used primarily with G.729 but could also be used with G.711. The default is None for G711 codecs and RFC2833 for G729.

Table 3-7Codecs and Fax Configuration Page Parameters

Parameter	Description
Priority	The Priority parameter defines the relative priorities to be offered during capabilities' exchange. If only G711A and G711U are used, the permitted priorities are 1 and 2. If all 3 codecs are used, the permitted priorities are 1, 2 and 3. Voice codec negotiation/priority is always performed between 2 end-points and depending on which side initiated the negotiation. The default is Priority 1 to G729, Priority 2 to G711A, and Priority 3 to G711U.
T38 Fax	The T38 check box indicates for each line whether to support the T38 Fax protocol. The default is checked (T38 Fax supported).

Click on the **Save** button before leaving the page to save the new settings. The new settings will be applied after restarting the unit.

3.7.1.2 Dial plan Schemes

A dial plan gives the unit a map to determine when a complete number has been entered and should be passed to the gatekeeper for resolution into an IP address. Dial plans are expressed using the same syntax as used by MGCP NCS specification. The following notation describes the formal syntax of the dial plan:

Digit ::= "0" | "1" | "2" | "3" | "4" | "5" | "6" | "7" | "8" | "9"

Timer ::= "T" | "t"

Letter ::= Digit | Timer | "#" | "*" | "A" | "a" | "B" | "b" | "C" | "c" | "D" | "d"

Range ::= "X" | "x" -- matches any digit

| "[" Letters "]" -- matches any of the specified letters

Letters::= Subrange | Subrange Letters

Subrange::= Letter -- matches the specified letter

| Digit "-" Digit -- matches any digit between first and last

Position::= Letter | Range

StringElement::= Position -- matches any occurrence of the position

| Position "." -- matches an arbitrary number of occurrences including 0

String ::= StringElement | StringElement String

StringList::= String | String "|" StringList

DialPlan::= String | "(" StringList ")"

A dial plan, according to this syntax, is defined either by a (case insensitive) string or by a list of strings. Regardless of the above syntax a timer is only allowed if it appears in the last position in a string (12T3 is not valid). Each string is an alternate numbering scheme. The unit will process the dial plan by comparing the current dial string against the dial plan. If the result is under-qualified (partial matches at least one entry) then it will do nothing further but wait until a full match is reached. If the result is over-qualified (no further digits could possibly produce a match) then it aborts the dial attempt and notifies end-user with an audio signal. Only a full match will trigger to initiate a call, by sending the dialed information to a Gatekeeper.

The Timer T is activated when it is all that is required to produce a match. The period of timer T is 4 seconds as default (configurable). For example a dial plan of (xxxT|xxxxx) will match immediately if any 5 digits are entered. It will also match following a 4 second pause after entering 3 digits.

Example 1 (simple dial plan):

Following example allows dialing any 7-digit number (e.g. 5551234) or an operator on 0. Dial plan is: (0T|xxxxxxx)

Example 2 (complex dial plan):

Local operator on 0, long distance operator on 00, four digit local extension number starting with 3,4 or 5, seven digit local numbers are prefixed by an 8, two digit star services (e.g. 69), ten digit long distance prefixed by 91, and international numbers starting with 9011+variable number of digits.
Dial plan for this is: (0T|00T|[3-5]xxx|8xxxxxxx|*xx|91xxxxxxxxxx|9011x.T)

Call completion

Call completion means allowing user to skip the timer period T after finished dialing, by ending number sequence with '#' (no other character is valid for this feature). A valid dial plan to accomplish this would be: (x.#|x.T)

Non-dialed Line Dial Plan

As soon as handset is lifted the unit contacts the gatekeeper (used for systems where DTMF detection is done in-call, i.e. "Hotline" model). Dial plan is then (x.), note the dot '.', which means match against 0 (or more) digits.

3.7.2 SIP Extensions Page

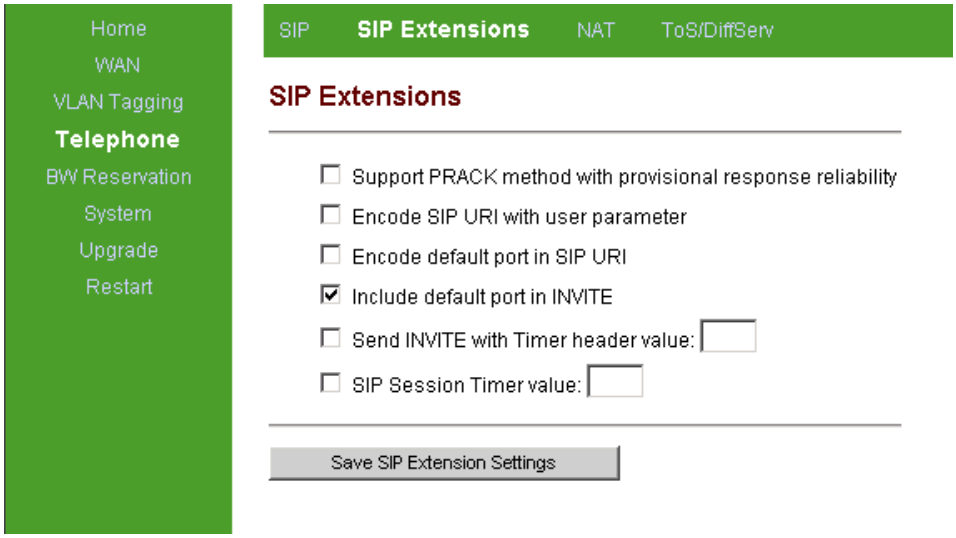

Figure 3-13: SIP Extensions Page

The SIP Extensions page includes the following components:

Table 3-8: SIP Extensions Page Parameters

Parameter	Description
Support PRACK method with provisional response reliability	The PRACK request plays the same role as ACK, but for provisional responses. PRACK is a normal SIP message, like BYE. As such, its own reliability is ensured hop-by-hop through each stateful proxy. Also like BYE, but unlike ACK, PRACK has its own response. If this were not the case, the PRACK message could not traverse proxy servers compliant to RFC 2543. For more details refer to RFC 3262: Reliability of Provisional Responses in the Session Initiation Protocol (SIP).
Encode SIP URI with user parameters	User=Phone will be inserted in the Contact field of SIP uniform resource identifier (URI).
Encode default port in SIP URI	Include default port in SIP uniform resource identifier (URI) even though it is not mandatory according to standard.
Include default port in INVITE	Include default port in the INVITE even though it is not mandatory according to standard

Table 3-8: SIP Extensions Page Parameters

Parameter	Description
Send INVITE with timer header value	If the called user agents (UA) or the SIP Proxy Server (SPS) requires a session timer for a requested session and the calling UA does not include the Session-Expires header in the INVITE message, then the called UA or the SPS may reject the request with a 487-request failure message. If the use of a session timer is desirable but optional for the session and the calling UA does not include the Session-Expires header in the INVITE then the called UA or SPS may add a Session-Expires header to the next session setup message. In this case, the SPS shall add the Session-Expires header to the INVITE message and the called UA shall add the Session-Expires header to the 200 OK response message. The range for the timer header value is from 1 to 999.
SIP Session timer value	The SIP Session Timer Support feature adds the capability to periodically refresh Session Initiation Protocol (SIP) sessions by sending repeated INVITE requests. The repeated INVITE requests, or re-INVITEs, are sent during an active call leg to allow user agents (UA) or proxies to determine the status of a SIP session. Without this keep alive mechanism, proxies that remember incoming and outgoing requests (stateful proxies) may continue to retain call state needlessly. If a UA fails to send a BYE message at the end of a session or if the BYE message is lost because of network problems, a stateful proxy does not know that the session has ended. The re-INVITES ensure that active sessions stay active and completed sessions are terminated. The range for the timer value is from 1 to 999 seconds.

Click on the **Save SIP Extensions Settings** button before leaving the page to save the new settings. The new settings will be applied after restarting the unit.

3.7.3 NAT Traversal Configuration Page

NAT Traversal function can be used to allow the Voice Gateway to register to a SIP proxy server even though the Voice Gateway is connected behind a NAT device.

Port forwarding may need to be activated for all telephone ports used by Voice Gateway: For example, RTP port range and SIP signaling ports.

The Keepalive timeout parameter in the Telephony page may also need to be set to a value lower than 1200 seconds to ensure that the Voice Gateway will not loose registration to the SIP server.

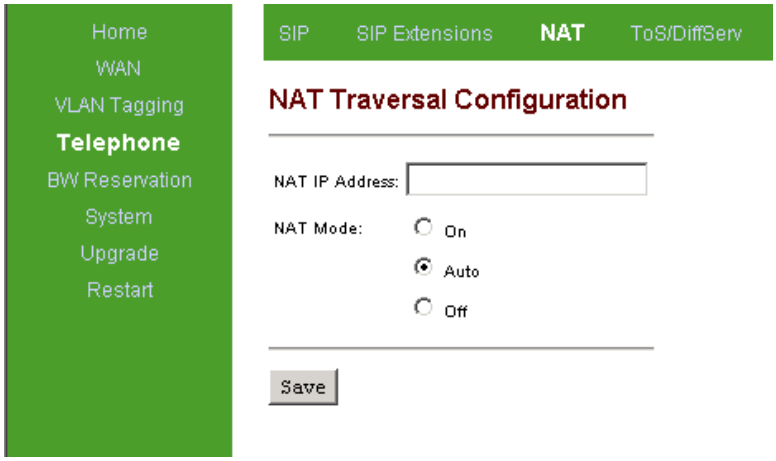

Figure 3-14: NAT Traversal Configuration Extensions Page

The NAT Traversal Configuration page includes the following components:

Table 3-9: NAT Traversal Configuration Page Parameters

Parameter	Description
NAT IP Address	The IP address that the NAT device uses on the WAN side. If the Voice Gateway is set to Auto NAT mode (see below), the IP address of the outside IP will be automatically inserted.
NAT Mode:	The NAT mode: ■ On = Enable NAT Traversal function using manual setting. ■ Auto = Enter NAT mode if any of the following conditions is met: a. IP-address = Private IP address b. "received" parameter in INVITE or REGISTER IP-address is not equal to internal IP address. ■ Off = NAT Traversal function is disabled. The default is Off.

Click on the **Save** button before leaving the page to save the new settings. The new settings will be applied after restarting the unit.

3.7.4 ToS/DiffServ Page

Outgoing packets from the Voice Gateway can be marked with DSCP (DiffServ Code Point) values. The ToS/DiffServ page enables defining the 8-bits ToS field in the IP header for different packet types. Diffserv use the first 6 out of these 8 bits. For more information about DiffServ Code Points please refer to RFC2474.

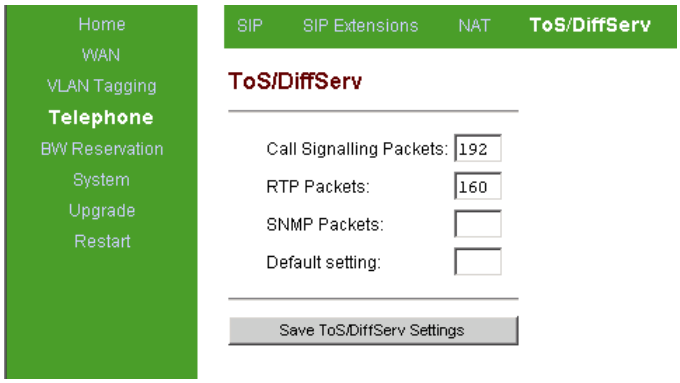

Figure 3-15: ToS/DiffServ Page

The ToS/DiffServ page includes the following components:

Table 3-10: ToS/DiffServ Page Parameters

Parameter	Description
Call signaling Packets	DiffServ marking for call signalling packets. Enter a number in the range 0 to 255 (The first 6 bits is the value of the DSCP field) or null. The default is 192.
RTP Packet	DiffServ marking for RTP and RTCP packets. Enter a number in the range 0 to 255 (The first 6 bits is the value of the DSCP field) or null. The default is 160.
SNMP Packets	DiffServ marking for SNMP packets. Enter a number in the range 0 to 255 (The first 6 bits is the value of the DSCP field) or null. The default is 0.
Default setting	DiffServ marking for other types of packets (e.g. HTTP, TFTP). Enter a number in the range 0 to 255 (The first 6 bits is the value of the DSCP field) or null. The default is 0.

Click on the **Save ToS/DiffServ Settings** button before leaving the page to save the new settings. The new settings will be applied after restarting the unit.

3.8 BW Reservation – DRAP Configuration Page

The Voice Gateway uses DRAP (Dynamic Resource Allocation Protocol) for efficient management of bandwidth resources for telephone calls.

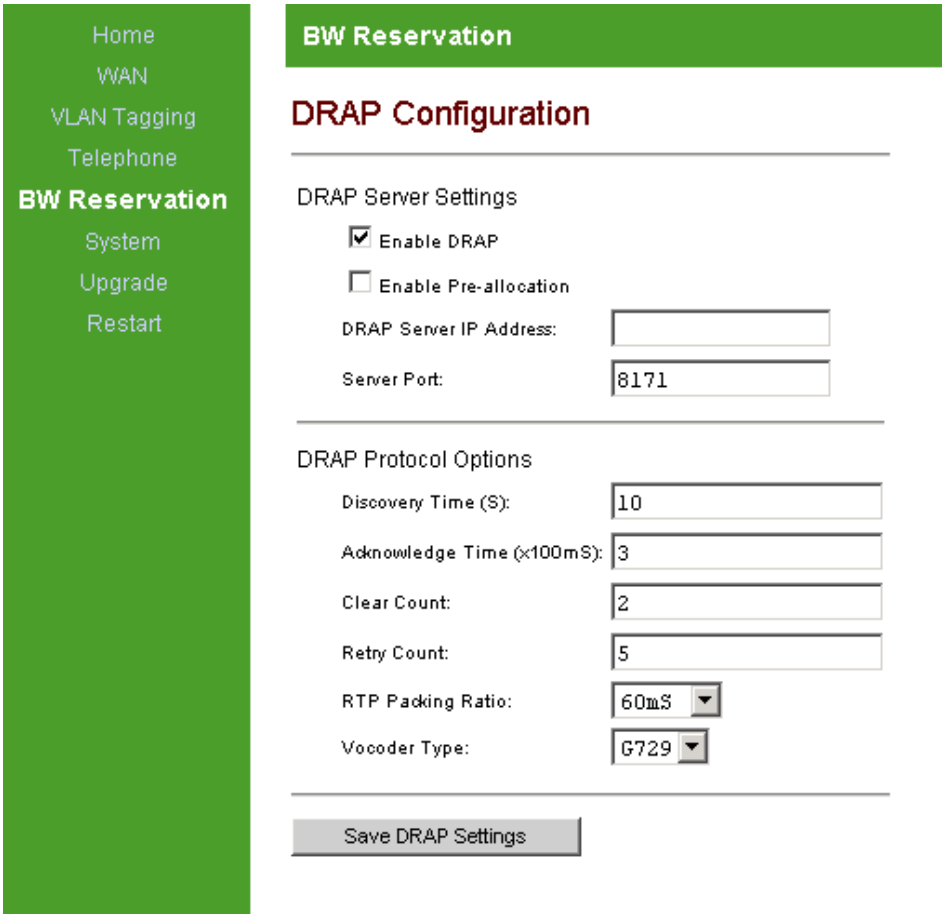

Figure 3-16: DRAP Configuration Page

The DRAP Configuration page includes the following components:

Table 3-11: DRAP Configuration Page Parameters

Parameter	Description
DRAP Server Settings	
Enable DRAP	The Enable DRAP option defines whether DRAP is used for establishing telephone (voice and fax) calls. If enabled, a DRAP Server must be available to provision telephone calls. The default is disabled (unchecked).

Table 3-11: DRAP Configuration Page Parameters

Parameter	Description
Enable Pre-allocation	The Enable Pre-allocation option defines whether resource allocation is requested immediately upon off-hook condition or only after dialing the requested number. When disabled (unchecked), a request for resource allocation will be sent only after dialing the number. When enabled, the resource allocation request will be sent immediately, and a dial tone will be provided only if the requested resources are available. The default is enabled (checked).
DRAP Server IP Address	The IP address of the DRAP server that should serve the resource allocation requests of the unit. Leave empty for Auto Discovery. The default is an empty field (Auto Discovery).
Server Port	The UDP port used for the DRAP server. The port number indicated will be used for originating ALLOC messages and the port number indicated +1 will be used for receiving CONFRM messages. The available range is from 8000 to 8200. The default is 8171.
DRAP Protocol Options	
Discovery Time	The Discovery Time is the timeout to be used when the Auto Discovery process is used for finding a DRAP server. The Auto Discovery process is based on sending empty broadcast allocation requests, and the Discovery Time is the time that the unit will wait for a response before sending a new request. The range is 1 to 255 seconds. The default is 10 seconds.
Acknowledge Time	The Acknowledge Time is the timeout out to be used between allocation requests. If no confirmation is received within this time, a new allocation request should be sent. The range is 1 to 10 (x 100 milliseconds). The default is 3 (300 milliseconds).

Table 3-11: DRAP Configuration Page Parameters

Parameter	Description
Clear Count	The Clear Count parameter indicates the number of allocation requests (ALLOC) that can be sent without being acknowledged before clearing all pending reservation attempts. Note: Established reservations (existing calls) are not cleared. The range is 1 to 10. The default is 2.
Retry Count	The Retry Count parameter indicates the number of allocation requests (ALLOC) that can be sent without being acknowledged before reaching a decision that the unit should search for another server. When this number is reached established reservations are to be cleared (existing calls are disconnected) and auto discovery procedure is initiated. The range is 1 to 10. The default is 5.
RTP Packing Ratio	The RTP Packing Ratio parameter defines the packet size to be used until an actual call is established. It is recommended to set a value that supports the worst-case scenario, e.g. the smallest expected size (20 milliseconds) that results in the highest expected number of packets per second. NOTE: The configured RTP Packing Ratio is used by the unit until an actual call is established. Once a call is established, the unit will use a packet size according to the actual value being used for the call. The available range is 10 to 100 milliseconds in multiples of 10 (10, 20, …100). The default value is 30 milliseconds.

Table 3-11: DRAP Configuration Page Parameters

Parameter	Description
Vocoder Type	The Vocoder Type parameter defines the codec to be used until an actual call is established. It is recommended to set a value that supports the worst-case scenario, e.g. the codec with the highest bandwidth requirement. Typically G711 should be configured, except in networks where only G729 is used. NOTE: The configured Vocoder Type is used by the unit until an actual call is established. Once a call is established, the unit will use the actual codec type being used for the call. The available options are G711 and G729. The default is G729.

Click on the **Save DRAP Settings** button before leaving the page to save the new settings. The new settings will be applied after restarting the unit.

3.9 System Menu

The System page selection bar includes the following options:

- Security (Set Security Password Page)
- Localization (Localization Page)
- SNMP (SNMP Configuration Page)
- Service Access (Service Access Configuration Page)

3.9.1 Set Security Password Page

Figure 3-17: Set Security Password Page

The Set Security Password page includes the following components:

Table 3-12: Set Security Password Page Parameters

Parameter	Description
New password	Enter the new password. A password includes up to 20 printable characters and is case sensitive. A null (empty) string means no password.
Confirm new password	Enter the new password again (must be the same as above).

Click on the **Save Password** button before leaving the page to save the new settings. The new settings will be applied after restarting the unit.

3.9.2 Localization Page

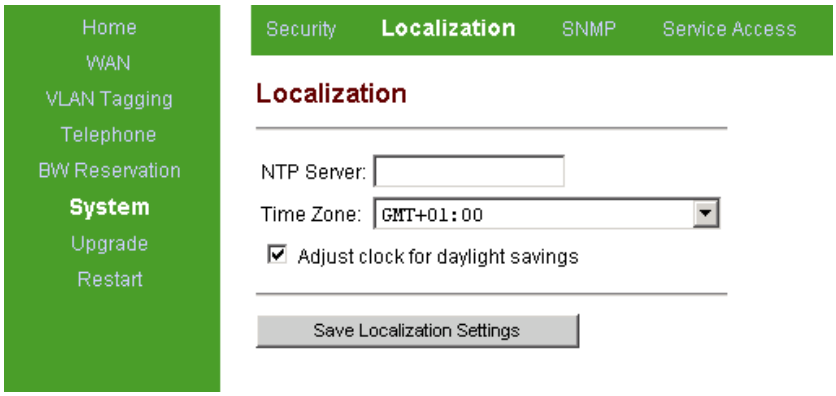

Figure 3-18: Localization Page

The Localization page includes the following components:

Table 3-13: Localization Page Parameters

Parameter	Description
NTP Server	The IP address of the NTP-server (optional). If an IP address is configured the NTP server usage is activated. The feature must be activated to support FSK-based caller ID. The default is disabled (no IP address).
Time Zone	The appropriate time zone. Use the drop-down list to change the time zone.
Adjust clock for daylight savings	By checking the “Adjust clock to daylight savings” the Voice Gateway will automatically adjust to daylight saving time (set the time one hour ahead). The default is enabled (checked).

Click on the **Save Localization Settings** button before leaving the page to save the new settings. The new settings will be applied after restarting the unit.

3.9.3 SNMP Configuration Page

Figure 3-19: SNMP Configuration Page

The SNMP Configuration page includes the following components:

Table 3-14: SNMP Configuration Page Parameters

Parameter	Description
SNMP Trap Configuration	
Trap Destination 1 to Trap destination 6	Specify up to 6 IP addresses to which SNMP traps should be sent. If all Trap Destinations are null, SNMP traps will be sent as broadcasts.
SNMP MIB Parameter Configuration	
Read Community	The read community string, up to 20 printable characters, case sensitive. Default string is public.
Write Community	The write community string, up to 20 printable characters, case sensitive. Default string is private

Click on the **Save SNMP Settings** button before leaving the page to save the new settings. The new settings will be applied after restarting the unit.

3.9.4 Service Access Configuration Page

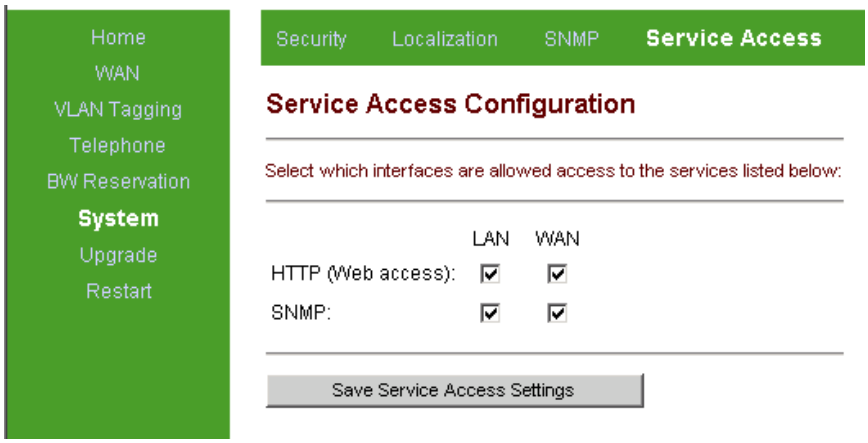

Figure 3-20: Service Access Configuration Page

The Service Access Configuration page enables to enable/disable access to various services. Access from each of the ports (LAN or WAN) using HTTP and/or SNMP can be either enabled or disabled. The default for all options is enabled (checked).

Click on the **Save Service Access Settings** button before leaving the page to save the new settings. The new settings will be applied after restarting the unit.

3.10 Upgrade Page

The Upgrade page enables to control the process of downloading either a software file (with the extension .ro) or a configuration file (with the extension .ini) from a TFTP-server.

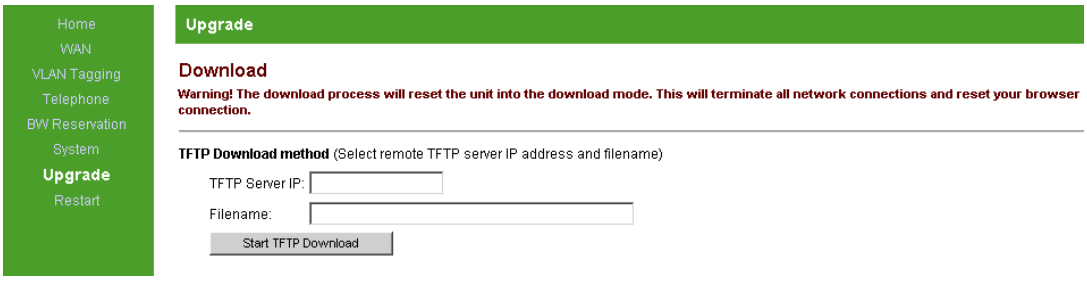

Figure 3-21: Upgrade Page

The Upgrade page includes the following components:

Table 3-15: Upgrade Page Parameters

Parameter	Description
TFTP Server	The IP address of the TFTP server
File name	The file name in the TFTP server of the software or the configuration .ini file. Up to 25 characters.

Click on the **Start TFTP Download** button to start the download process. The downloading and installation of the new SW version or configuration file is done automatically, including a restart of the unit. When the installation is complete and the unit has restarted, the Home Product Info page will be displayed.

3.10.1 Downloader Result Codes (hexadecimal)

If something goes wrong during download or installation, you will be notified according to the following:

- 0 (0x00): normal boot (no upgrade requested or needed)
- bit-0 (0x01): upgrade requested or main application not valid
- bit-1 (0x02): failed to download new image
- bit-2 (0x04): TFTP server not defined
- bit-3 (0x08): TFTP file not defined

- bit-4 (0x10): TFTP session failed
- bit-5 (0x20): CRC error in downloaded image
- bit-6 (0x40): incompatible image

3.10.1.1 Examples

- An attempt to download from a non-existing TFTP-server results in code 0x7 (= 0x07):
  - bit-2 0x04 TFTP server not defined plus…
  - bit-1 0x02 failed to download new image plus…
  - bit-0 0x01 upgrade requested or main application not valid
- An attempt to download a non-existing file results in code 0xb (= 0x0b):
  - bit-3 0x08 TFTP file not defined plus…
  - bit-1 0x02 failed to download new image plus…
  - bit-0 0x01 upgrade requested or main application not valid
- A successful download results in code 0x01
- A restart without download of main application results in 0x00.

3.11 Restart Page

When settings have been inserted or altered, the Voice Gateway must be restarted in order to apply the new settings.

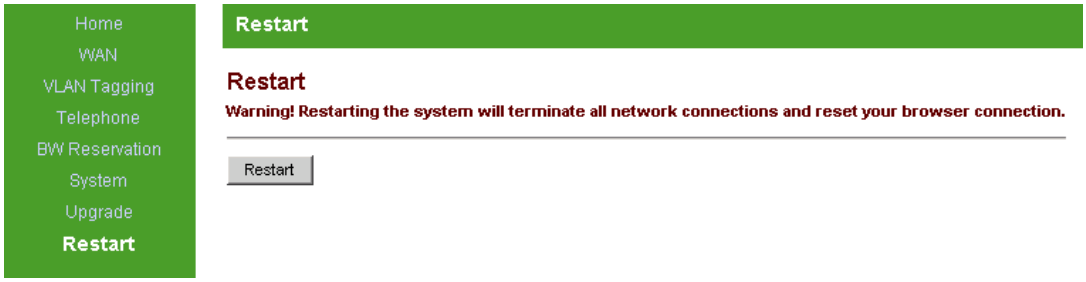

Figure 3-22: Restart Page

Click the **Restart** button to restart the Voice Gateway.

3.12 Parameters Summary

Table 3-16: Parameters Summary

Parameter	Range/Options	Default
WAN Configuration Page		
Device Operating Mode	Only Bridge option is available	Bridge
Obtain WAN configuration dynamically	Yes (checked)/No (unchecked)	Yes (selected)
Client identity	Standard/Custom The Custom string can include up to 25 characters	Standard The default Custom string is null
Vendor ID	A string of up to 25 characters	VoIP (used by default but is not displayed)
Specify fixed WAN configuration	Yes (checked)/No (unchecked)	No (unchecked)
IP Address	IP address	192.168.254.254
Subnet Mask	IP address	255.255.255.0
Default Gateway	IP address	Null
DNS Address	IP address	Null
Host Name	A string of up to 25 characters	Null
Domain Name	A string of up to 25 characters	Null
Broadcast Limit	0%-100%	100%
Multicast Limit	0%-100%	100%
VoIP VLAN Configuration Page		
Call Signaling VLAN Tag	1-4094 or null	Null
Call Signaling Priority Tag	0-7 or null	Null
RTP VLAN Tag	1-4094 or null	Null
RTP Priority Tag	0-7 or null	Null

Table 3-16: Parameters Summary

Parameter	Range/Options	Default
SIP Configuration Page		
Dialplan	A string of up to 100 characters. For details on format see section 3.7.1.2	x.T|x.#
Dial Timeout	1-60 seconds	4 seconds
Adaptive Jitter Buffer Maximum Duration	100-300 milliseconds	100 milliseconds
RTP Port Range	Start/Stop: 1030-65535	Start: 8000 End: 8015
Telephone Line	On/Off	Off
HA Mode	Fixed/Auto/Off	Off
SIP Server IP (primary)	IP address	Null
SIP Server Port (primary)	1030-65535	5060
SIP Server IP (secondary)	IP address	Null
SIP Server Port (secondary)	1030-65535	5060
User Name	A string of up to 25 characters	Null
Password	A string of up to 25 characters	Null
Outgoing Display Name	A string of up to 25 characters	Null
Telephone number	A string of up to 25 characters	Null
Telephone domain name	A string of up to 25 characters	Null
Port	1030-65535	Line 1: 5060 Line 2: 5061
Incoming CLIP	On/Off	Off
Keepalive timeout	10-65535 seconds	1200 seconds
Ring signal	0-9	0
Transport	UDP/TCP	UDP

Table 3-16: Parameters Summary

Parameter		Range/Options	Default
Codecs and Fax Configuration			
G711A	Select/Deselect	Yes (checked)/No(unchecked)	Always selected
	SS	Not applicable	
	EC	Enable/Disable	Enabled
	Packet	10-150 milliseconds	20 ms
	Keypad	None, RFC2833 or SIP INFO	None
	Priority	1-3	2
G711U	Select/Deselect	Yes (checked)/No(unchecked)	Always selected
	SS	Not applicable	
	EC	Enable/Disable	Enabled
	Packet	10-150 milliseconds	20 ms
	Keypad	None, RFC2833 or SIP INFO	None
	Priority	1-3	3
G729	Select/Deselect	Yes (checked)/No(unchecked)	Yes (checked)
	SS	Enable/Disable	Disabled
	EC	Enable/Disable	Enabled
	Packet	10-150 milliseconds	30 ms
	Keypad	None, RFC2833 or SIP INFO	RFC2833
	Priority	1-3	1
T38 Fax		Enable/Disable	Enable
SIP Extensions Page			
Support PRACK method with provisional response reliability		Yes/No	No (unchecked)
Encode SIP URI with user parameters		Yes/No	No (unchecked)

Table 3-16: Parameters Summary

Parameter	Range/Options	Default
Encode default port in SIP URI	Yes/No	No (unchecked)
Include default port in INVITE	Yes/No	Yes (checked)
Send INVITE with timer header value	Yes/No, plus a value in the range 1-999 seconds if Yes (checked) is selected.	No (unchecked) The default value is null
SIP Session timer value	Yes/No, plus a value in the range 1-999 seconds if Yes (checked) is selected.	No (unchecked) The default value is null
NAT Traversal Configuration Page		
NAT IP Address	IP address	Null
NAT Mode	On/Auto/Off	Off
ToS/Diffserv Page		
Call signaling Packets	0-255 or null	192
RTP Packets	0-255 or null	160
SNMP Packets	0-255 or null	0
Default setting	0-255 or null	0
DRAP Configuration Page		
Enable DRAP	Enable/Disable	Disable (unchecked)
Enable Pre-allocation	Enable/Disable	Enable (checked)
DRAP Server IP Address	IP address or null for auto discovery	Null (auto discovery)
Server Port	8000-8200	8171
Discovery Time	1-255 seconds	10 seconds
Acknowledge Time	1 to 10 (x 100 milliseconds)	3 (300 milliseconds)
Clear Count	1-10	2
Retry Count	1-10	5

Table 3-16: Parameters Summary

Parameter	Range/Options	Default
RTP Packing Ratio	10 to 100 milliseconds in multiples of 10 (10, 20, …100)	30 milliseconds
Vocoder Type	G711/G729	G729
Set security Password Page		
New password/ Confirm new password	Up to 20 printable characters, case sensitive. A null (empty) string means no password.	
Localization Page		
NTP Server	IP address or null for disable NTP server	Null (NTP server disabled)
Time Zone	Drop Down Menu	GMT+01:00
Adjust clock for daylight savings	Yes/No	Yes (checked)
SNMP Configuration Page		
Trap Destination 1 to Trap destination 6	IP addresses. If all Trap Destinations are null, SNMP traps will be sent as broadcasts.	Null for all addresses
Read Community	Up to 20 printable characters, case sensitive.	public
Write Community	Up to 20 printable characters, case sensitive.	private
Service Access Configuration Page		
HTTP LAN	Yes/No	Yes (checked)
HTTP WAN	Yes/No	Yes (checked)
SNMP LAN	Yes/No	Yes (checked)
SNMP WAN	Yes/No	Yes (checked)

Table 3-16: Parameters Summary

Parameter	Range/Options	Default
Upgrade Page		
TFTP Server	IP address	
File name	A string of up to 25 characters	

A

Appendix A - Internal Class 5 Services

In This Appendix:

This appendix provides a description of the internal Class 5 services that are supported by the Voice Gateway.

A.1 Actions and Keypad Sequences

Keypad Sequences		
Action	**Description**	**Keypad Sequence (R=hook-flash)**
HOLD	Holds an on-going call	R0
DROP	Drops an on-going call	R1
FLASH	Switches between on-going call sessions or starts new call inquiry	R2
CONFERENCE	Activates 3-party conference	R3
CONFERENCE DROP	Deactivates 3-party conference	R5
CW ACTIVATION	Enables Call Waiting indication tone	*43#
CW DEACTIVATION	Disable Call Waiting indication tone	#43#
CW STATUS CHECK	Informs about the current configuration of Call Waiting indication tone	*#43#
CALL FORWARD ACTIVATION	Call Forward activation	*21* <telephone number>#
CALL FORWARD DEACTIVATION	Call Forward de-activation	#21#

A.2 Using the Class 5 Services

A.2.1 Call Waiting

Description: One on-going call active, audible CW tone indicating new incoming call in progress.

Call Waiting Service	
Action	**Event**
R0	Reject incoming call -> Calling party hears busy tone. Continue with active call.
R1	Disconnect on-going call and answer incoming call.
R2	Place on-going call on hold, answer incoming call.

A.2.2 Call Inquiry

Description: One on-going call active, place a new call to a third party.

Call Inquiry Service	
Action	**Event**
R2+telephone number	Place on-going call on hold (dial tone), Inquire new call to a third party.
R1	Return to call placed on hold if third party is not answering.

A.2.3 Call Alteration

Description: Two on-going calls active, switch between calls.

Call Alteration Service	
Action	**Event**
R2	Switch between two on-going calls. Places non-active call on hold.

A.2.4 Call Drop

Description: Two on-going calls active, disconnect one of the calls.

Call Drop Service	
Action	**Event**
R0	Disconnect call that is put on hold. Continue with on-going call.
R1	Disconnect on-going call and return to call that is put on hold.

A.2.5 3-Party Conference 1

Description: One on-going call active, place a new call to a third party and start conference.

3-Party Conference Service 1	
Action	**Event**
R3+telephone number	Place on-going call on hold (dial tone), inquire new call to a third party and mix all session into a conference when third party has answered.
R5	End conference with third party and return to first initiated call session.

A.2.6 3-Party Conference 2

Description: Two on-going calls active, mix them into a conference session.

3-Party Conference Service 2	
Action	**Event**
R3	Start conference with all active parties (mix audio streams).
R5	End conference with third party and return to first initiated call session.

A.2.7 Call Waiting Indication Tone

Description: Available only when there are no calls active/in progress.

Call Waiting Indication Tone Service	
Action	**Event**
*43#	Enable Call Waiting indication tone
#43#	Disable Call Waiting indication tone (calling party will hear a busy tone when calling)
*#43#	Informs about present Call Waiting indication tone configuration: ■ Three short beeps = off ■ Two long beeps = on

A.2.8 Call Forward

Description: Available only when there are no calls active/in progress.

Call Forward Service	
Action	**Event**
21<telephone number>#	Enable Call Forward and do forward to <telephone number>. Indication tone is heard.
#21#	Deactivate Call Forward

B

Appendix B - Default Telephony Parameters

In This Appendix:

This appendix provides the default settings for various telephony parameters.

Table B-1: Default Telephony Parameters

Parameter	Definition	Default
Normal Ringing Signal	The signal that end user will hear from the telephone set when a call is received	Cadence: 1 second on, 4 second off Duration: 180 seconds Frequency: 25 Hz
Ringing Tone	The tone that sounds on the telephone set when ringing on the other side.	Cadence: 1 second on, 5 second off Duration: Not limited Frequency: 425 Hz Level: -10 dBmO
Dial Tone	The tone that the call originator hears in the handset before dialing the destination telephone number.	Cadence: Continuous Duration: Not limited Frequency: 425 Hz Level: -5 dBmO
Busy Tone	The tone that the end user that originates a call hears when the destination telephone line is busy.	Cadence: 0.25 second on, 0.25 second off Duration: Not limited Frequency: 425 Hz Level: -10 dBmO
Network Busy Tone	The tone that the end user that originates a call will hear when the network is congested.	Cadence: 0.25 second on, 0.75 second off Duration: Not limited Frequency: 425 Hz Level: -10 dBmO
Call Waiting Tone	The tone that the end user that originates the call hears when there is a second incoming call during the original call session.	Cadence: 0.2 second on, 0.5 second off, 0.2 seconds on Duration: One On-Off-On cycle Frequency: 425 Hz Level: -10 dBmO

Table B-1: Default Telephony Parameters

Parameter	Definition	Default
Caller ID Standard	The country standard and data transmission method	Sweden, DTMF
CID Alerting Method	The method of alerting on the existence of CID data	Polarity Reversal
On-hook data transmission timing method	The timing for transmission of CID data	Post first ring signal
2-Wire Impedance	The impedance presented between the A and B wires of the telephone line in active state (nominal impedance).	CTR21 (ETSI complex = 270Ω+750Ω||150nF))

Glossary

BW	Band Width
CLIP	Calling Line Identification Presentation: A supplementary service used to show the number of a caller.
CNI	Calling Number Identification.
DHCP	Dynamic Host Configuration Protocol. A protocol for dynamically assigning IP addresses from a pre-defined list to nodes on a network. Using DHCP to manage IP addresses simplifies client configuration and efficiently utilizes IP addresses.
DNS	Domain Name System: The name resolution system that lets users locate computers on the Internet (TCP/IP network) by domain name. The DNS server maintains a database of domain names (host names) and their corresponding IP addresses.
DRAP	Dynamic Resource Allocation Protocol
DSCP	Differentiated Service Code Point, AKA DiffServ: An alternate use for the ToS byte in IP packets. Six bits of this byte are being reallocated for use as the DSCP field where each DSCP specifies a particular per-hop behavior that is applied to the packet.
DTMF	Dual-Tone Multi Frequency: The type of audio signals that are generated when you press the buttons on a touch-tone telephone. DTMF assigns a specific frequency (consisting of two separate tones) to each key.
EC	Echo Cancellation
EMC	Electro-Magnetic Compatibility. The capability of equipment or systems to be used in their intended environment within designed efficiency levels without causing or receiving degradation due to unintentional EMI (Electro Magnetic Interference). EMC generally encompasses all of the electromagnetic disciplines.

FSK Frequency Shift Keying: A simple modulation technique that merges binary data into a carrier. It creates only two changes in frequency: one for 0, another for 1.

G.711 A 64 Kbps PCM voice-coding technique. Described in the ITU-T standard in its G-series recommendations.

G.729 A compression technique where voice is coded into 8 Kbps streams. There are two variations of this standard (G.729 and G.729 Annex A) that differ mainly in computational complexity; both provide speech quality similar to 32-kbps ADPCM. Described in the ITU-T standard in its G-series recommendations.

H.225 An ITU standard protocol for control signaling in an H.323 audio or video environment. H.225 uses messages defined in H.245 to establish the call over the Registration, Admission and Signaling (RAS) channel.

H.245 An ITU standard protocol for control messages in an H.225 audio or videoconferencing call. The messages are used for flow control, encryption and jitter management as well as for initiating the call, negotiating which features should be used and terminating the call. It also determines which side is the master for issuing various commands.

H.323 A protocol suite defined by ITU-T for voice transmission over internet (Voice over IP or VoIP). In addition to voice applications, H.323 provides mechanisms for video communication and data collaboration, in combination with the ITU-T T.120 series standards.

HA High Availability.

HTTP HyperText Transport Protocol: A protocol used to request and transmit files, especially web pages and web page components, over the Internet or other computer network.

IDU Indoor Unit

IEC International Electrotechnical Commission, Geneva, Switzerland, www.iec.ch: An organization that sets international electrical and electronics standards founded in 1906. It is made up of national committees from over 60 countries.

IEEE Institute of Electrical and Electronics Engineers. IEEE (pronounced I-triple-E) is an organization composed of engineers, scientists, and students. The IEEE is best known for developing standards for the computer and electronics industry. In particular, the IEEE 802 standards for local-area networks are widely followed.

IEEE 802.1p A QoS method - A three-bit value that can be placed inside an 802.1Q frame tag.

IEEE 802.1Q The IEEE 802.1Q standard defines the operation of VLAN Bridges that permit the definition, operation and administration of Virtual LAN topologies within a Bridged LAN infrastructure. The 802.1Q specification establishes a standard method for inserting VLAN membership information into Ethernet frames. A tag field containing VLAN (and/or 802.1p priority) information can be inserted into an Ethernet frame, carrying VLAN membership information.

IETF Internet Engineering Task Force. One of the task forces of the IAB (Internet Architecture Board), formally called the Internet Activities Board, which is the technical body that oversees the development of the Internet suite of protocols (commonly referred to as "TCP/IP").The IETF is responsible for solving short-term engineering needs of the Internet.

IP Internet Protocol. The standard that defines how data is transmitted over the Internet. IP bundles data, including e-mail, faxes, voice calls and messages, and other types, into "packets", in order to transmit it over public and private networks.

ISO International Organization for Standardization, Geneva, www.iso.ch: An organization that sets international standards, founded in 1946. The U.S. member body is ANSI. ISO deals with all fields except electrical and electronics, which is governed by the older International Electrotechnical Commission (IEC). With regard to information processing, ISO and IEC created JTC1, the Joint Technical Committee for information technology.

ITU-T International Telecommunication Union – Telecommunications. An intergovernmental organization through which public and private organizations develop telecommunications. The ITU was founded in 1865 and became a United Nations agency in 1947. It is responsible for adopting international treaties, regulations and standards governing telecommunications. The standardization functions were formerly performed by a group within the ITU called CCITT, but after a 1992 reorganization the CCITT no longer exists as a separate entity.

LAN Local area Network. A computer network limited to a small geographical area, such as a single building. The network typically links PCs as well as shared resources such as printers.

MAC Media Access Control. The lower of the two sub-layers of the data link layer defined by the IEEE. The MAC sub-layer handles access to shared media, such as whether token passing or contention will be used.

MAC Address Standardized data link layer address that is required for every port or device that connects to a LAN. Other devices in the network use these addresses to locate specific ports in the network and to create and update routing tables and data structures. MAC addresses are 6bytes long and are controlled by the IEEE.

MIB Management Information Base. A database of objects that can be monitored by a network management system. SNMP uses standardized MIB formats that allow any SNMP tools to monitor any device defined by a MIB.

NAT Network Address Translation: An IETF standard that allows an organization to present itself to the Internet with far fewer IP addresses than there are nodes on its internal network. The NAT technology, which is typically implemented in a router, converts private IP addresses (such as in the 192.168.0.0 range) of the machine on the internal private network to one or more public IP addresses for the Internet. It changes the packet headers to the new address and keeps track of each session. When packets come back from the Internet, NAT performs the reverse conversion to the IP address of the client machine.

NTP Network Time Protocol: A protocol used to update the real time clock in a computer. There are numerous primary and secondary servers in the Internet that are synchronized to the Coordinated Universal Time (UTC) via radio, satellite or modem. For more information, visit www.ntp.org.

POTS Plain Old Telephone System. A basic analog telephone equipment.

PPPoE Point-to-Point Protocol over Ethernet. PPPoE relies on two widely accepted standards: PPP and Ethernet. PPPoE is a specification for connecting the users on an Ethernet to the Internet through a common broadband medium, such as a single DSL line, wireless device or cable modem. All the users over the Ethernet share a common connection, so the Ethernet principles supporting multiple users in a LAN combines with the principles of PPP, which apply to serial connections.

RTP Real Time Protocol. An Internet protocol for transmitting real-time data such as audio and video. RTP itself does not guarantee real-time delivery of data, but it does provide mechanisms for the sending and receiving applications to support streaming data. Typically, RTP runs on top of the UDP protocol, although the specification is general enough to support other transport protocols.

SIP Session Initiation Protocol. An application-layer control IETF protocol that can establish, modify, and terminate multimedia sessions such as Internet telephony calls (VoIP). SIP can also invite participants to already existing sessions, such as multicast conferences. Media can be added to (and removed from) an existing session. SIP transparently supports name mapping and redirection services, which supports personal mobility - users can maintain a single externally visible identifier regardless of their network location.

SNMP Simple Network Management Protocol. A network management protocol that provides a means to monitor and control network devices, and to manage configurations, statistics collection, performance, and security. SNMP works by sending messages, called protocol data units (PDUs), to different parts of a network. SNMP-compliant devices, called agents, store data about themselves in Management Information Bases (MIBs) and return this data to the SNMP requesters.

SS Silence Suppression: A method for eliminating wasted bandwidth when sending voice over a packet-switched system.

SU Subscriber Unit

T.38 This is the ITU-T recommendation which defines a real time method for fax over IP networks.

TCP Transmission Control Protocol. Connection-oriented transport layer protocol that provides reliable full-duplex data transmission. TCP is the part of the TCP/IP suite of protocols that is responsible for forming data connections between nodes that are reliable, as opposed to IP, which is connectionless and unreliable.

TCP/IP Transmission Control Protocol/Internet Protocol. A set of protocols developed by the U.S. Department of Defense to allow communication between dissimilar networks and systems over long distances. TCP/IP is the de facto standard for data transmission over networks, including the Internet.

TFTP Trivial File Transfer Protocol. Simplified version of FTP that allows files to be transferred from one computer to another over a network, usually without the use of client authentication.

ToS Type Of Service: A field in an IP packet (IP datagram) that is used for quality of service (QoS).

UDP User Datagram Protocol. Connectionless transport layer protocol in the TCP/IP protocol stack. UDP is a simple protocol that exchanges datagrams without acknowledgments or guaranteed delivery, requiring that error processing and retransmission be handled by other protocols. UDP is defined in RFC 768.

V.17 An ITU fax standard (1991) that uses TCM (Trellis-Coded Modulation, a technique for forward error correction) modulation at 12,000 and 14,400 bps for Group 3 faxes. It adds TCM to the V.29 standard at 7,200 and 9,600 bps to allow transmission over noisier lines. It also defines special functions (echo protection, turn-off sequences, etc.) for half-duplex operation. Modulation use is a half-duplex version of V.32bis.

V.29 An ITU standard (1976) for synchronous 4,800, 7,200 and 9,600 bps full-duplex modems using QAM (Quadrature Amplitude Modulation) on four-wire leased lines. It has been adapted for Group 3 fax transmission over dial-up lines at 9,600 and 7,200 bps.

V.32bis An ITU standard (1991) for asynchronous and synchronous 4,800, 7,200, 9,600, 12,000 and 14,400 bps full-duplex modems using TCM and echo cancellation. It supports rate renegotiation, which allows modems to change speeds as required.

VLAN Virtual Local Area Network. A group of devices on one or more LANs that are configured with the same VLAN ID so that they can communicate as if they were attached to the same wire, when in fact they are located on a number of different LAN segments. Used also to create separation between different user groups.

VoIP Voice over Internet Protocol. Provides an advanced digital communications network that bypasses the traditional public switched telephone system and uses the Internet to transmit voice communication. VoIP enables people to use the Internet as the transmission medium for telephone calls by sending voice data in packets using IP rather than by traditional circuit switched transmissions of the PSTN.

WAN Wide Area Network. A computer network that spans a relatively large geographical area. Wide area networks can be made up of interconnected smaller networks spread throughout a building, a state, or the entire globe.